AMENDMENTS TO THE CLAIMS

The following listing of claims will replace all prior versions and listings of claims in the application.

LISTING OF CLAIMS

1. (Original) A method of performing speaker adaptation upon speech models associated with a speech recognizer, the speech models having been created under first environmental conditions, comprising:

obtaining input speech under second environmental conditions from a speaker for whom the speech models are to be adapted and extracting observation data from said input speech;

decoding said observation data to ascertain state segmentation data associated with said observation data;

providing a linear approximation operator that embeds a priori knowledge of said first environmental conditions;

operating upon said observation data using said linear approximation operator and said state segmentation data to transform said observation data into compensated observation data that approximates the observation data under said first environmental conditions;

applying a speaker adaptation operation upon said compensated observation data to generate adapted speech models for said speaker.

2. (Original) The method of claim 1 wherein said observation data is extracted by performing feature extraction upon said input speech.

3. (Original) The method of claim 1 wherein said observation data is extracted by generating cepstral coefficients based on said input speech.

4. (Original) The method of claim 1 further comprising:

determining the difference between said first and second environmental conditions;

using said difference and said linear approximation operator to alter said speech models to at least approximately match said second environmental conditions; and

using said altered speech models to perform said decoding step.

5. (Original) The method of claim 1 wherein said operating step is performed by applying the inverse of said linear approximation operator to said observation data.

6. (Original) The method of claim 4 wherein said step of altering said speech models prior to decoding is performed by applying said linear approximation operator to said speech models; and

wherein said operating step is performed by applying the inverse of said linear approximation operator to said observation data.

7. (Original) The method of claim 1 wherein said adaptation operation is commutative with said linear approximation operator.

8. (Original) The method of claim 1 wherein said adaptation operation employs maximum a posteriori estimation.

9. (Original) The method of claim 1 wherein said adaptation operation employs maximum likelihood linear regression.

10. (Original) The method of claim 1 wherein said linear approximation operator employs a Jacobian matrix.

11. (Original) The method of claim 1 wherein said linear approximation operator employs a Jacobian matrix modified by a linear transformation.

12. (Original) A speaker adaptation system comprising:

a speech recognizer employing a first set of speech models created under first environmental conditions, said speech recognizer having an input through which a user provides input speech under second environmental conditions and having an output that supplies observation data corresponding to said second environmental conditions;

a speaker adaptation module coupled to said speech recognizer, said speaker adaptation module performing a speaker adaptation process upon said first set of speech models based on said observation data;

said speaker adaptation module further having linear approximation operator that stores knowledge of said first environmental conditions and adjusts said observation data to correspond to said first environmental conditions and thereby compensate for differences between said first and second environmental conditions.

13. (Original) The adaptation system of claim 12 wherein said speech recognizer employs a feature extraction component to develop said observation data.

14. (Original) The adaptation system of claim 12 wherein said recognizer employs a decoder for providing state segmentation information to said speaker adaptation module.

15. (Original) The adaptation system of claim 12 wherein said observation data are cepstral coefficients.

16. (Original) The adaptation system of claim 12 wherein said speech recognizer is a noise compensated recognizer.

17. (Original) The adaptation system of claim 12 wherein said speech recognizer employs a feature extraction component to develop a measure of the difference between said first and second environmental conditions.

18. (Original) The adaptation system of claim 17 further comprising a second linear approximation operator operable to adjust said first set of speech models based on said measure of the difference between said first and second environmental conditions.

19. (Original) The adaptation system of claim 12 wherein said adaptation module employs maximum a posteriori estimation.

20. (Original) The method of claim 12 wherein said adaptation module employs maximum likelihood linear regression.

21. (Original) The method of claim 12 wherein said linear approximation operator employs a Jacobian matrix.

22. (Original) The method of claim 12 wherein said linear approximation operator employs a Jacobian matrix modified by a linear transformation.

23. (New) The method of claim 2, further comprising utilizing a special linear approximation of background noise that is applied after feature extraction and prior to speaker adaptation to allow the speaker adaptation to adapt speech models to an enrolling user without distortion from the background noise.

24. (New) The method of claim 5, wherein the inverse linear approximation operator operates upon said observation data, using the state segmentation data, and a result of the inverse approximation is a set of modified observation data that has been cleaned up to remove effects of background noise, and speaker adaptation is then performed.

25. (New) The method of claim 24, wherein the speaker independent acoustic models, once adapted, are used to extract even more accurate state segmentation, which is then used to perform an even more precise inverse linear approximation operation with further improved speaker adaptation.